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Source-Adaptation-Based Wireless Video Transport: A Cross-Layer Approach

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Real-time packet video transmission over wireless networks is expected to experience bursty packet losses that can cause substantial degradation to the transmitted video quality. In wireless networks, channel state information is hard to obtain in a reliable and timely manner due to the rapid change of wireless environments. However, the source motion information is always available and can be obtained easily and accurately from video sequences. Therefore, in this paper, we propose a novel cross-layer framework that exploits only the motion information inherent in video sequences and efficiently combines a packetization scheme, a cross-layer forward error correction (FEC)-based unequal error protection (UEP) scheme, an intracoding rate selection scheme as well as a novel intraframe interleaving scheme. Our objective and subjective results demonstrate that the proposed approach is very effective in dealing with the bursty packet losses occurring on wireless networks without incurring any additional implementation complexity or delay. Thus, the simplicity of our proposed system has important implications for the implementation of a practical real-time video transmission system.

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1. INTRODUCTION

The characteristics of wireless channels provide a major challenge for reliable transport of real-time multimedia applications since the data transmitted over wireless channels are highly sensitive to the noise, interference, and the multipath environment that can cause both packet loss and bit errors. Furthermore, these errors tend to occur in bursts, which can further decrease the delivered quality of service (QoS) [1–3]. Current and future 3G systems will have to cope with this lack of QoS guarantees. As a result, the need exists for video coding and transmission schemes that not only provide efficient compression performance, but also provide relatively robust transport performance in the presence of link errors resulting in bursty packet losses.

The issue of supporting error-resilient video transmission over error-prone wireless networks has received considerable attention. A number of techniques have been proposed to combat the effects of packet losses over wireless networks and thereby increase the robustness of the transmitted video [4]. In [5, 6], a “smart” inter/intramode switching scheme is proposed based on an RD analysis, but the effectiveness of this approach with bursty packet losses is not clear and it may be too complicated for implementation in real-time video applications. In [7], a model-based packet

interleaving scheme is studied that can provide some performance gain at the cost of additional delay since the interleaving is spread over several video frames; thus, this scheme is not appropriate for real-time video applications due to the relatively large delay induced. In [2, 8–10], the effect of different forward error correction (FEC) coding schemes on reconstructed video quality has been investigated. The use of FEC-based unequal error protection (UEP) is considered as an effective tool in dealing with channel errors since it can provide different levels of protection to different classes of data that can be classified based on their relative importance to reconstructed image quality. In this way, system resources, such as bandwidth, can be utilized efficiently. In [11], a bit-level UEP approach is proposed; however, most of today’s networks are packet oriented and thus in [12, 13], packet-level UEP approaches based on the relative data importance are investigated. However, in [11, 13], the proposed systems have not considered the use of the characteristics of the video content and they only considered FEC coding alone as an error-resilience technique. As indicated in [12, 14], the error-resilience techniques should be efficiently combined and the video content should be seriously considered in choosing the protection redundancy of the transmitted video. More specifically, [14] describes a media-dependent FEC algorithm relying on an MPEG-2

syntactic structuring technique and a judicious combination of protection redundancy, MPEG syntactic data and pure video information are shown to greatly improve the video quality under a given bit-rate budget. In [12], it has been shown that the video motion information is an important factor that can determine the appropriate protection level in face of time-varying source/channel dynamics and has led to an efficient system combining multiple error-resilience techniques while exploiting the source/channel dynamics. However, in wireless networks channel state information is hard to obtain in a reliable and timely manner due to the rapid change of wireless environments and in many scenarios, such as video multicasting and broadcasting, this feedback information is completely unavailable. Therefore, in such scenarios it is difficult to adapt to the channel conditions since the unreliable channel feedback will substantially degrade the system performance.

Therefore, as discussed above, we do not consider adaptation to channel conditions based on feedback information from the destination. Instead, we focus on adaptation to source motion information since this information is always available to the encoder and can easily be communicated to the decoder(s). Based on this observation, we propose a novel framework that efficiently combines multiple error-resilience techniques, that is, a robust packetization scheme, a motion-based FEC/UEP scheme, a motion-based intracoding rate selection scheme as well as a novel intraframe interleaving.

More specially, in this work we explore a source-adaptive cross-layer FEC/UEP scheme based on the motion information extracted from a video sequence to be encoded. This approach is based on the notion that, for a given video frame, the loss of high-motion portions can cause relatively larger distortion compared to other lower-motion portions due to the increased perceptual importance of this high-motion information [15, 16]. Clearly, we then need to protect the high-motion portion with stronger FEC coding, while weaker FEC protection should suffice for the less significant low-motion portion. In this paper, we consider an H.264 encoder/decoder and take the level of motion associated with a slice¹ as an indication of the relative importance of the corresponding data. The motion levels associated with a slice are classified in terms of the mean-square values of the corresponding interframe prediction errors. We then use different Reed-Solomon (RS) codes to protect the slice depending on the computed interframe motion levels thereby achieving UEP. In order to facilitate the FEC/UEP approach, a novel packetization scheme based on the universal mobile telecommunication system (UMTS) protocol architecture [17] is proposed, which can simultaneously provide efficient source coding performance and robust delivery. Furthermore, this approach does not induce any additional delay when used together with the proposed FEC/UEP scheme compared to traditional packetization schemes.

Clearly, the robustness provided by intracoding comes at some expense, as it generally requires a higher bit rate than more efficient intercoding schemes to achieve the same reconstructed video quality. So how to balance the error robustness achieved by intracoding with the resulting reduction in source coding efficiency is an important issue. In this framework, we also include a source-adaptive intracoding rate selection scheme that is based on exponential weighted moving average (EWMA) estimation of the local motion level. Using this scheme, an appropriate intracoding rate is selected for each group of N successive frames based on an estimate of the corresponding relative motion level of those N successive frames.

Finally, for the purpose of real-time video transmission, we make use of an intraframe interleaving scheme that interleaves the video/parity packets within a frame. Thus, since the delay is constrained within a single video frame, no additional delay is incurred, while this scheme is still capable of substantially randomizing the burst losses occurring on wireless networks. Therefore, improved performance can be expected.

The contributions or novelties of this paper consist of (1) providing a robust video coding and transmission framework for scenarios where channel feedback is not available or cannot be obtained easily or accurately; (2) exploiting the characteristics of video source content to adaptively select the protection level in terms of intracoding rate and channel coding rate; (3) efficiently combining multiple error-resilience techniques to optimize the system performance. Furthermore, the packet losses in previous related work [2, 3] are modeled at the network layer for wireless IP networks using the RTP/UDP/IP protocol stack, and no effort is made to model the packet losses at the link layer. A further contribution of the present paper is extending this previous work by explicitly modeling the packet losses at the link layer taking into account the effects of packet segmentation that takes place at this layer.

The rest of the paper is organized as follows. In Section 2, we discuss the proposed framework that efficiently combines a packetization scheme, a cross-layer motion-based FEC/UEP approach, a source-adaptive intracoding rate selection scheme, and a novel intraframe interleaving scheme and at the end of this section we discuss the computational complexity and the standards compliance of this proposed approach. Simulation results are presented in Section 3 followed by conclusions in Section 4.

2. PROPOSED ERROR-RESILIENCE FRAMEWORK

Motivated by the discussion in the previous section, we propose a cross-layer wireless video transport framework that efficiently combines multiple error-resilience techniques for scenarios where the channel state information is not available or cannot be easily or accurately obtained. Instead, the system only adapts to source motion information inherent in the source video sequences. In what follows, we first describe the components of the proposed framework and then discuss

¹ A slice, in general, consists of a selected number of macroblocks; in this work, it is defined as a whole horizontal row of macroblocks.

the computational complexity and the standards compliance issues of this approach.

2.1. Proposed packetization scheme

The introduction of slices in the H.264 encoding process has at least two beneficial aspects for video transmission over wireless networks. The two primary factors are the reduced error probability of smaller packets and the ability to resynchronize within a frame [17]. However, the use of slices also adversely affects the source coding efficiency due to the increased slice overhead and reduced prediction accuracy within one frame, since interframe motion vector prediction and intraframe spatial prediction are not allowed across slice boundaries in H.264 [17]. Therefore, on one hand, the number of slices in a frame should not be too large (small slice size). Despite the improved resynchronization capabilities associated with small slice sizes, the increased overhead information would compromise source coding efficiency. On the other hand, the number should not be too small (large slice size), due to the higher error probabilities associated with the larger slice sizes. In [2, 17], it is demonstrated that using 6–9 slices per QCIF frame is a reasonable choice for a wide range of operating bit rates and channel conditions. Of course, in face of error-free transmission this choice would be worse than using 1 slice per frame due to the drop in source coding efficiency. However, as shown in [1, 17] this choice would achieve a better tradeoff between source coding efficiency and error resilience in a wide range of realistic channel conditions. For details, please refer to [1, 17].

When transmitting over the wireless network, the application layer video packets are further segmented into radio link protocol (RLP) packets at the link layer. This segmentation can cause some problems. If the existing transport protocol is TCP, unless all the RLP packets belonging to the same TCP packet are received successfully within the retransmission limit set at the link layer, the entire application packet will be discarded. On the other hand, if the existing transport protocol is UDP, unless all the RLP packets are received successfully, the entire application packet will also be lost unless the UDP error checking feature is disabled. Nevertheless, UDP has other desirable properties compared to TCP for real-time video transport applications. Thus, we will concentrate on the use of UDP with error checking disabled² as the transport protocol in this paper.

Based on the discussion above, our proposed packetization approach is implemented as follows.

(1) In the encoding process, every slice in a frame consists of an equal number of MBs (in this paper, we exclusively use 11 MBs per slice, thus every QCIF video frame is divided into 9 slices). Then, every encoded slice is packetized into one RTP/UDP/IP packet, which is also called an application packet.

(2) Since the induced packet overhead for every RTP/UDP/IP packet is 40 bytes, in order to economize on the scarce bandwidth resource, we use robust header compression (ROHC) [18] to compress the RTP/UDP/IP header into 3 bytes with no UDP checksums set. Then, the overhead of packet data convergence protocol (PDCP) is attached.

(3) At the link layer, every application packet is divided into k equal-sized RLP packets according to the associated maximum transmission unit (MTU) of this transmission. The value of k is kept constant for the whole transmission session. We add a header to every RLP packet, one part of which can be used to allow the FEC decoder to determine the positions of lost packets and the other part can be used to indicate which FEC code is used for the FEC/UEP scheme as will be discussed later. Generally, the header size is determined by the segmentation procedure at the link layer and the number of different RS codes used for the FEC scheme. In this paper, we use a 5-bit header, 3 bits to determine the position information, and the remaining 2 bits to indicate which RS code is employed.³

(4) Then, the proposed FEC/UEP scheme in Section 2.2 is applied to the set of RLP packets that belong to the same application packet. The data packets, together with the parity packets, are then delivered over the network.

(5) Finally, at the receiver, the FEC decoder first recovers the lost packets and if every RLP packet within an application packet is received correctly, the corresponding application packet is delivered to the upper layer; if not, the corresponding application packet is discarded.

Based on this proposed scheme, it is possible to achieve improved source coding efficiency and relative robustness compared to other approaches, such as using a large number of slices in one frame. The process is illustrated in Figure 1, for example, for 3G UMTS wireless networks.

Since our approach is on a link-by-link basis, we model the packet loss process for video delivery at the link-layer level instead of at the network layer. The loss model we use is the two-state Gilbert model [19] illustrated in Figure 2, which can be uniquely specified by the average burst length (L_B) and the packet loss rate (P_L). They can be related to the corresponding state transition probabilities p and q according to $p = P_L/L_B(1 - P_L)$ and $q = 1/L_B$.

2.2. FEC-based unequal error protection: a cross-layer scheme

In this subsection, based on the proposed packetization scheme we introduce the novel cross-layer FEC-based UEP approach.

2.2.1. Priority classification

In previous work [15], motion/activity has been determined by the interframe prediction error that is used as a primary indicator of activity of video frames. Also, in [12, 16, 20],

² The reason for disabling the error checking capability is that, in this paper, we have not included the link-layer ARQ mechanism into the cross-layer framework. So, in order to fairly evaluate the performance of the proposed approach, we need to disable the link-layer retransmission function.

³ As a result, we assume that no more than four codes are employed in the FEC/UEP scheme.

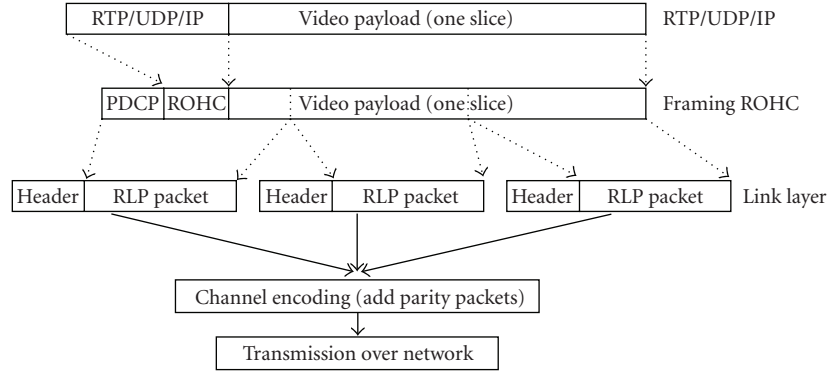


FIGURE 1: Proposed packetization for UMTS wireless networks.

the interframe prediction error of a frame is used to determine its motion/activity level. The results in [12, 16, 20] have demonstrated that this simple way to determine the motion or activity level of video frames is effective. Therefore, we will also use this statistical classifier in our motion-based adaptive system to classify the motion levels of slices.

For the transmitted video sequence, at the application layer we first calculate the mean-square prediction error between slices that are in the same position in successive frames according to

$$E[m, n] = \sum_{j=0}^{N_v-1} \sum_{i=0}^{N_h-1} [X_{m,n}(i, j) - X_{m-1,n}(i, j)]^2, \quad (1)$$

where $E[m, n]$ denotes the mean-square prediction error between the luminance data in the n th slice of size $N_v \times N_h$ pixels of the m th frame, and the corresponding data in the n th slice of the $(m-1)$ th frame of the video sequence where the total number of frames is N_f . Here, $X_{m,n}(i, j)$ represent the luminance values at pixel position (i, j) in the n th slice of frame m .

In Figure 3, the measured slice prediction errors are indicated for the QCIF Foreman and the QCIF Susie sequences at 10 fps and 30 fps, respectively. The results are perfectly consistent with subjective observations for these two sequences. That is, the Susie sequence is considered to have low motion with the background tending to remain constant. On the other hand, the Foreman sequence has much more motion due to increased activity and scene changes. Also, for each sequence some portions of frames (slices) can be seen to have more interframe motion than others.

Our FEC/UEP scheme is based on the slice prediction errors computed at the application layer. As illustrated in Figure 3, we set two thresholds T_1 and T_2 that are different for each sequence and are chosen as indicated later. We classify the slices in a video sequence into one of the following three motion levels.

High-priority class: slice prediction error above T_1 .

Medium-priority class: slice prediction error between T_1 and T_2 .

Low-priority class: slice prediction error below T_2 .

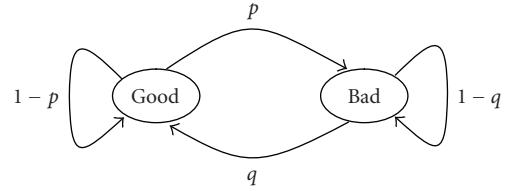


FIGURE 2: Two-state Gilbert model.

2.2.2. Unequal error protection interleaved Reed-Solomon coding

In our approach, UEP is realized by assigning an unequal amount of FEC at the link layer to classes with different motion levels. This is a cross-layer approach since the link layer requires the motion information obtained at the application layer to implement the FEC/UEP scheme.

For each slice in the same class, the same interleaved RS code is applied across the RLP packets after the corresponding application packet is split into k equal size RLP packets,⁴ that is, for k RLP packets in the same application packet, an interleaved RS(n, k) code is applied with $k \leq n$. In this paper, we use interleaved RS encoding as described in [21]. Afterwards, the resulting n packets are transmitted over a UMTS wireless network. At the receiver side, after the k RLP packets for a single application packet are received, the FEC decoder can identify the positions of lost packets using the header information of each RLP packet. The lost packets can be recovered if the number of lost packets is less than $n - k$. If any one of the RLP packets cannot be recovered, the entire application packet is discarded.

Thus, since different classes use an unequal amount of redundancy, UEP is achieved. By using this approach, slices with higher motion are protected by stronger RS codes, while slices with less motion are protected by weaker RS

⁴ In this paper, we choose $k = 3$ as a compromise between the introduced overhead and source coding efficiency.

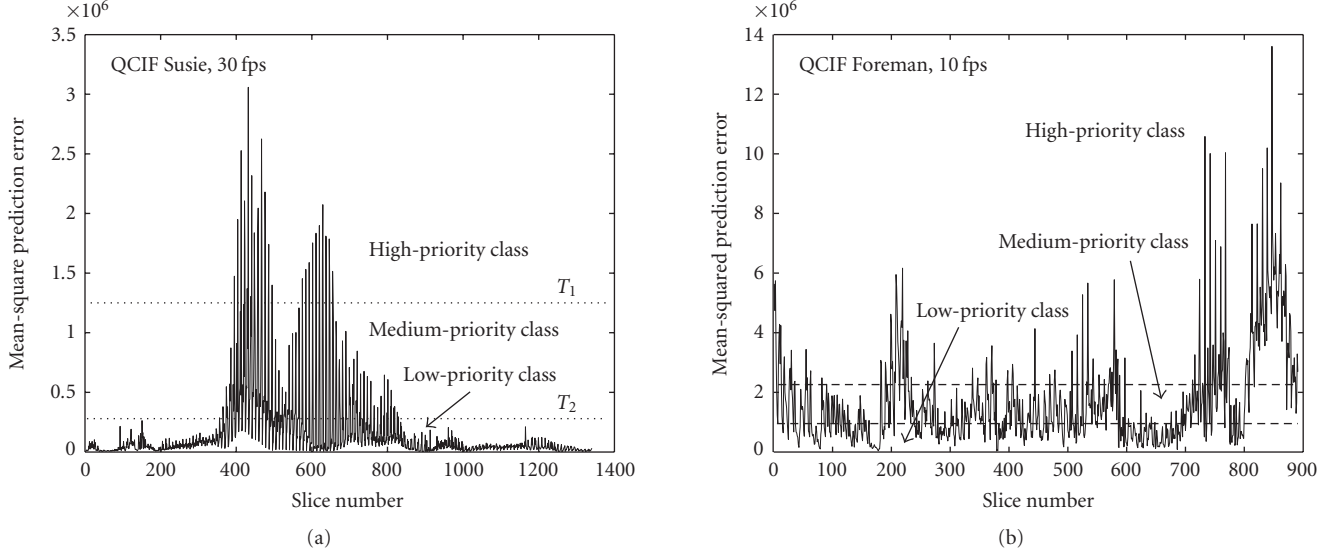


FIGURE 3: Slice prediction error for both QCIF Foreman and QCIF Susie sequence.

codes. In this way, the system bandwidth resources can be utilized more efficiently. Actually, the loss of a low-motion frame/slice is barely noticeable since it can be effectively concealed by the built-in passive error concealment capabilities used together with intraupdating. However, without FEC the loss of a high-motion frame/slice may cause substantial performance degradation in reconstructed video quality, especially when severe error propagation is considered [12].

We should also note that in this approach, the FEC coding is applied across the RLP packets associated with a single application packet. As a result, there is no noticeable delay introduced by this FEC/UEP approach. That is because, at the encoder side, after the application packets within one frame are segmented into several RLP packets, we can buffer the RLP packets while simultaneously sending the RLP packets to the receiver. The buffered RLP packets can then be used in the FEC encoding process to compute the parity packets so that no need exists to delay the transmission of the RLP data packets. Likewise, at the receiver side, a decoding delay is incurred only for those application packets with lost RLP packets, otherwise no delay is induced.

When implementing this FEC/UEP system, a practical issue arises in code application. More specifically, since different FEC codes are used, the receiver requires information about which RS code has been applied in order to decode the received packets. As indicated previously, the receiver can be notified of this information through a specified field in the RLP packet header.

2.3. Source-adaptive intracoding rate selection

Intracoding has been recognized as an important approach to constrain the effect of packet loss for motion-compensated based-video coding schemes. In this paper we study the effectiveness of a source-adaptive low-complexity intracoding scheme instead of the “smart” RD-based approach as

TABLE 1: Intraupdating rates employed.

Medium intraupdating	1 intraupdated slice/2 frames
High intraupdating	1 intraupdated slice/1 frames

described in [6]. In this scheme, one slice in every N frames is intracoded to enhance the error resilience in the face of packet losses. The specific intraupdating rates⁵ used in this paper are summarized in Table 1. In the absence of packet losses, the use of intraupdating can degrade the source coding efficiency. However, in the presence of packet loss, performance gains are expected due to the resulting improved error-resilience although, as we demonstrate, this depends on the degree of motion present in the source material. Therefore, a source-adaptive intracoding rate selection approach is proposed based on the estimated motion level associated with a video sequence.

In order to facilitate the encoding and transmission of real-time video content, in this work we make use of an exponential weighted moving average (EWMA) approach to make use of the information of the past frames as well as the current frame to estimate the average motion of N adjacent video frames in order to select the appropriate intraupdating rate for the current N frames.

In Figure 4, we illustrate the process of estimating the motion associated with $N = 2$ contiguous frames.⁶ In Step 1, we estimate the mean-square prediction error for the

⁵ The number of intraupdating rates can obviously be arbitrary although we consider only two rates in this paper which is sufficient to demonstrate the efficacy of the proposed source-adaptive approach.

⁶ In the results provided in what follows, for consistency with the intraupdating rates listed in Table 1, we make exclusive use of $N = 2$, although N can be arbitrary.

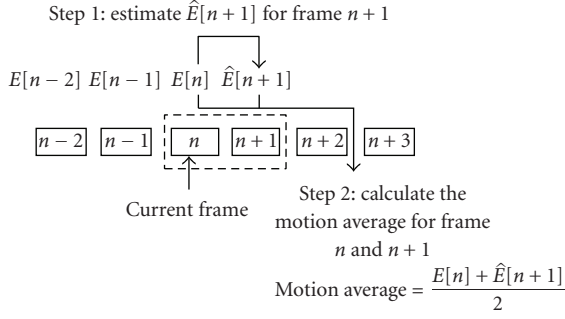


FIGURE 4: EWMA-based source-adaptive intraupdating rate selection.

$(n+1)$ th frame, denoted as $\hat{E}[n+1]$ in Figure 4, based on the actual mean-square prediction error and the estimated mean-square prediction error of the current frame according to

$$\hat{E}[n+1] = (1 - \alpha)\hat{E}[n] + \alpha E[n], \quad (2)$$

with $\hat{E}[0] = E[0]$. Here, $E[n]$ is the mean-square prediction error of the n th frame and $\hat{E}[n]$ is the estimated version for the same frame that has a memory that includes all of the past video frames; α is a weighting factor. Therefore, we can obtain

$$\hat{E}[n+1] = (1 - \alpha)^n E[0] + \alpha \sum_{i=1}^n (1 - \alpha)^{n-i} E[i], \quad (3)$$

which illustrates why this method is called an exponential weighted moving average (EWMA) estimate, that is, the estimated prediction error for the $(n+1)$ th frame can be expressed in terms of the prediction errors of all the past frames with exponentially decreasing weights. In this paper, the weighting parameter α is selected to be equal to 0.75.

After we obtain the estimated mean-square prediction error for the $(n+1)$ th frame, in Step 2, we calculate the motion average (MA) for the n th and $(n+1)$ th frame according to

$$MA = \frac{E[n] + \hat{E}[n+1]}{2} = \frac{(1 + \alpha)E[n] + (1 - \alpha)\hat{E}[n]}{2}, \quad (4)$$

or equivalently,

$$MA = \frac{1 + \alpha}{2} E[n] + \frac{\alpha(1 - \alpha)}{2} E[n-1] + \frac{\alpha(1 - \alpha)^2}{2} E[n-2] + O((1 - \alpha)^2). \quad (5)$$

Since we have $\alpha = 0.75$, the MA of the current two frames can be approximated as

$$MA \approx \frac{7}{8} E[n] + \frac{3}{32} E[n-1] + \frac{3}{128} E[n-2]. \quad (6)$$

Therefore, only the current frame and the 2 most recent frames contribute to estimating the motion average for the current $N = 2$ contiguous frames. This motion average is

then used by the video encoder to select an appropriate intraupdating rate for the current $N = 2$ successive frames. Then the n th and $(n+1)$ th frames are encoded with the selected intraupdating rate. The same process will take place for the subsequent $(n+2)$ th and $(n+3)$ th frames.

In Figure 5, we show the estimated motion average versus the actual motion average for every two consecutive frames in the QCIF Susie and Foreman sequences. From this figure, it can be seen that the EWMA-based motion estimation approach is quite accurate for sequences with considerably different levels of motions.

The calculated motion average for each $N = 2$ consecutive frames will be classified into two different classes: low motion and high motion. Based on this classification, we select the high intraupdating rate for high-motion frames and the medium intraupdating rate for low-motion frames. Thus, a source-adaptive intraupdating rate selection scheme is achieved that simultaneously takes into account the error-resilience requirements for video streams with different motion levels and the associated source coding efficiency. The adaptation logic implemented at the encoder is based on the use of a prestored threshold that can classify video frames into either a low-motion or high-motion class. This threshold has been obtained empirically and is used to instruct the video encoder what operation to be performed based on the current motion information. The threshold employed here to classify frames into high-motion class and low-motion class is identical to that used in [15]. Furthermore, it has been shown in [12] that a single threshold is sufficient to provide an appropriate classification and the use of finer thresholds would not enhance the system performance very much, especially considering the introduced complexity.

2.4. Intraframe interleaving

Since in wireless networks, packet losses tend to occur in bursts, which can cause serious performance degradation, it is desirable to randomize the burst packet losses. For traditional interleaving schemes, this is generally achieved by interleaving the application packets over several successive frames [7]; thus, a large delay is introduced. This is not practical for real-time video applications. In this paper, we propose an intraframe interleaving scheme that is distinct from traditional schemes. In this scheme, interleaving is only applied to the packets within the same video frame. Therefore, unlike traditional schemes, no extra delay is introduced.

Generally, interleaving can be implemented at two different layers: at the link layer or at the application layer. The application-layer interleaving is transparent to the underlying transport network. Therefore, it is readily applicable to a wide range of networks without any special requirements. The link layer interleaving approach, however, is a cross-layer design. Since the link layer has to obtain specific information from the upper layers to construct link-layer packets. Therefore, it is applicable on a link-by-link basis. In the case of a UMTS network, the connections between communicating parties are logical links. Therefore, both application-layer and link-layer interleaving schemes are applicable.

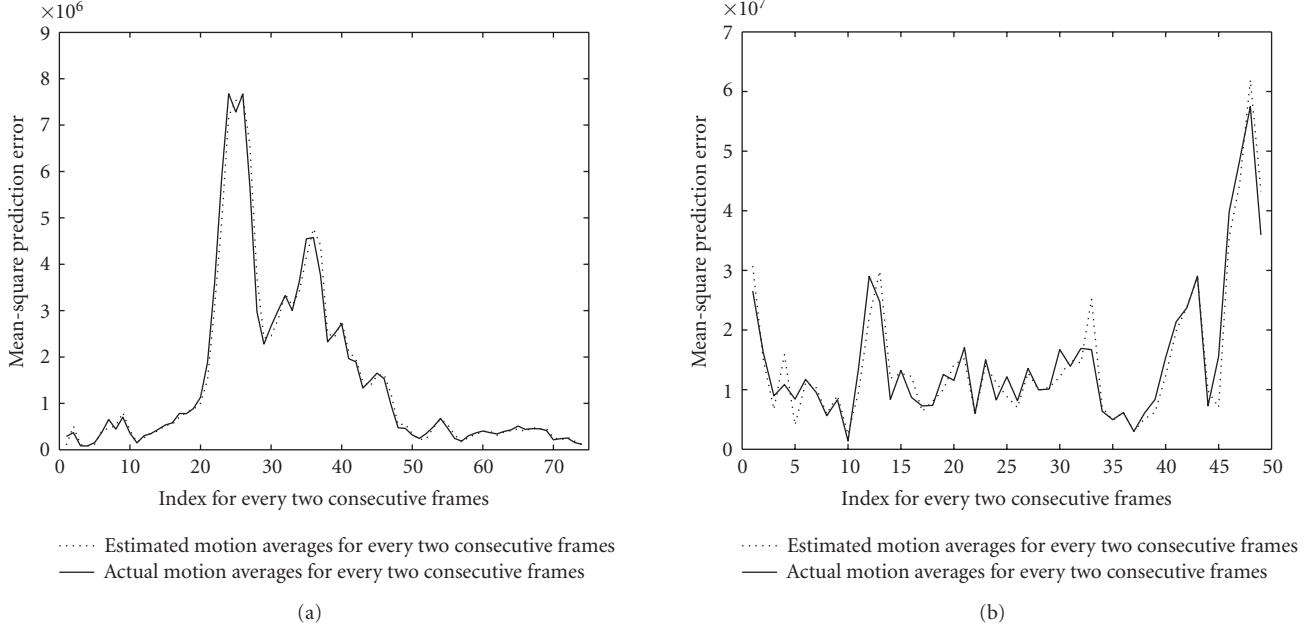


FIGURE 5: Estimated versus actual motion averages for two contiguous frames for QCIF Susie (a) and Foreman (b) sequences.

In this work, under the scope of UMTS networks, we have implemented the proposed intraframe interleaving scheme at each of the two layers and provide a performance comparison between the two different implementation methods in Section 3.

2.4.1. Application-layer intraframe interleaving

For the application-layer interleaving, we only interleave the positions of slices within a given video frame at the application layer. In cases where two or more successive slices are lost due to burst errors occurring on the wireless networks, this application-layer intraframe interleaving can help to maintain the effectiveness of the built-in passive error concealment (PEC) algorithm [22] by randomizing slice-level burst losses since successful reception of the neighboring slices is important for high performance recovery of any single lost slice.

In Figure 6, we illustrate the application-layer interleaving scheme. Since in this paper each frame is split into 9 slices, we interleave the positions of the nine slices within the frame according to the pattern illustrated in Figure 6. As can be seen in Figure 6, if we assume no interleaving and a burst loss of length 3 affecting slice#0, slice#1, and slice#2, then at the decoder side it is difficult for the motion-based PEC scheme to conceal the effects of this burst loss since no neighboring MBs are available for slice#0 and slice#1, and for slice#2 only slice#3 is available. Therefore, the performance of the PEC scheme is degraded significantly. However, if application-layer interleaving is applied, then for the same burst loss, after interleaving and subsequent deinterleaving, the burst loss is randomized to some extent as illustrated in Figure 6. In this case the performance of the PEC scheme

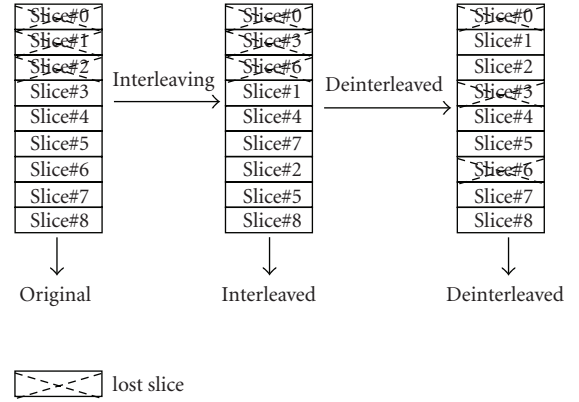


FIGURE 6: Application-layer intraframe interleaving scheme.

can be significantly improved since the necessary information for effective operation of the PEC algorithm is available from neighboring MBs.

2.4.2. Link-layer intraframe interleaving

In this approach, we employ an intraframe-based link-layer interleaving scheme that interleaves the RLP packets within a single frame instead of slices at the application layer.

In Figure 7, we illustrate this interleaving scheme. Since in this paper each frame is split into nine slices, and then at the link layer each slice is divided into three RLP packets plus appropriate parity packets based on the particular FEC/UEP scheme employed, we interleave the positions of the RLP packets within the frame according to the pattern illustrated in Figure 7. After we receive the interleaved link-layer RLP

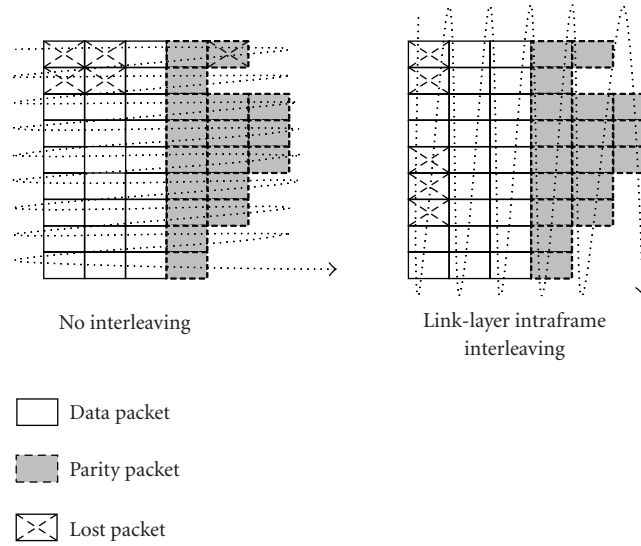


FIGURE 7: Link-layer intraframe interleaving scheme.

packets at the receiver, we deinterleave them, and then perform channel decoding. Thus, the burst errors at the link layer can be substantially randomized and, therefore, can result in improved effectiveness of the FEC scheme. In Figure 7, indicating the RLP packets within a video frame, if we assume there is an error pattern as indicated, when no interleaving is applied we have 5 packets lost during transmission. After channel decoding, this results in the first 2 slices lost since the number of lost packets exceeds the error correcting capabilities of the interlaced RS codes that are applied to each slice. However, when the link-layer intraframe interleaving scheme is employed within the video frame, the lost packets are redistributed and thus, in this case, all the losses can be corrected through RS channel decoding resulting in no lost slices. Therefore, a substantial performance gain can be achieved through joint use of FEC/UEP and the proposed link-layer intraframe interleaving scheme.

2.5. Computational complexity and standard compliance

We first discuss the computational complexity of the proposed approach and then the standard compliance. As can be seen, the major computational complexity resides in the intracoding rate selection and the FEC/UEP coding scheme. The computational complexity of the proposed intracoding rate selection is much less than the one proposed in [6]. More specifically, in [6] the inter/intracode switching is based on a rate-distortion (RD) framework where it is required to compute the distortion and two moments for the luminance value of each pixel for the cases of intracoding and intercoding. As shown in [6], for every pixel in an intercoded MB, 16 addition/16 multiplication operations are required and for each pixel in an intracoded MB, 11 addition/11 multiplication operations are required. By comparison, in the proposed intracoding rate selection scheme,

regardless of the MB coding mode, we need only 3 addition/4 multiplication operations for each pixel that is a substantial reduction in computational complexity. Furthermore, in the FEC/UEP scheme since the prediction error has already been computed for the intracoding selection, the only operation in the FEC encoder is a threshold comparison that can be seen as 1 multiplication. Thus the overall computational burden in the proposed system is substantially alleviated. Therefore, for real-time applications, our approach is able to provide an appropriate framework for providing adaptive intracoding rate selection as well as FEC/UEP coding that simultaneously considers the source coding efficiency and error-resilience behavior.

As for the standards compliance of this approach, with representative video coding standards, firstly in order to facilitate the motion-based intracoding rate selection and the FEC/UEP coding schemes, at the video encoder we need to compute the prediction error of each pixel, then this motion information is employed in the video encoder to determine an appropriate intracoding rate and is employed in the FEC encoder to implement the FEC/UEP operation. This latter operation requires only a minor change of a typical encoder protocol stack since it is a cross-layer scheme where the motion information should be delivered to the link layer where the FEC/UEP is implemented. Secondly, for the packetization scheme the only change is that we segment each application packet into equal-size link-layer packets and this can be achieved by setting the appropriate parameter in the link layer. Thirdly, in order to deinterleave at the receiver, necessary information encapsulated in the packet header should be provided to the decoder. Therefore, we can see that the changes to a standard video communication system are minor and easy to implement. Finally, although in this paper we make use of the H.264 coding standard to demonstrate the efficacy of the proposed approach, it is generally applicable to any other coding standard, such as MPEG-4, since the

proposed framework does not have any specific requirements on the video source encoder.

3. SIMULATION RESULTS AND DISCUSSIONS

This section presents simulation results to demonstrate the potential performance gain that can be achieved by the proposed error-resilience framework for packet video transport over 3G UMTS wireless networks.

Video sequences are encoded using the ITU-JVT JM codec [23] of the newly developed H.264 video coding standard. In this paper, we will use two typical QCIF test video sequences: Foreman and Susie, as described previously. The Foreman sequence at 10 fps is regarded as a high-motion-level sequence while Susie at 30 fps is regarded as a low-motion-level sequence. Both are coded at constant bit rates specified by using the associated rate control scheme [24]. The first frame of the sequence is intracoded and the rest of the frames are intercoded as P frames with adaptive intraupdating rate selection for each $N = 2$ contiguous frames. In our packetization scheme, each slice is packetized into one application packet, thus every QCIF frame is packetized into 9 application packets.

For the motion-based FEC/UEP scheme, an RS(6, 3) code is used for the high-priority class, an RS(5, 3) code is used for the medium-priority class, and an RS(4, 3) code is employed for the low-priority class. For comparison, we also investigate the performance of an equal error protection (EEP) scheme without interleaving; here the packetization process is the same as the UEP case, but we use a fixed RS(5, 3) code for all classes. The thresholds T_1 and T_2 indicated previously are then chosen so that the overall channel coding rates of these two systems are approximately equal. It should be noted that the link-layer retransmission function is disabled so that we only consider the use of FEC coding for recovery of lost packets.

In the simulation results, we also compare the performance of the proposed link-layer intraframe interleaving with application-layer intraframe interleaving. As described previously, application-layer intraframe interleaving only interleaves the positions of application packets (slices) in order to make the built-in error concealment more effective. On the other hand, the link-layer interleaving is intended to improve the effectiveness of the FEC/UEP approach.

The simulation results presented in Figures 8–11 are obtained using EWMA-based estimation for adaptive intraupdating rate selection together with the proposed motion-based FEC/UEP scheme. We also present a comparison between EWMA-based adaptive intraupdating rate selection and use of fixed intraupdating rates in Figures 12 and 13.

In Figure 8, we show the results for the Foreman sequence for the case of burst length $L_B = 3$, where we choose the thresholds T_1 and T_2 such that out of a total of 900 application packets, 205 application packets are classified into the high-priority class, 500 packets into the medium-priority class, and 195 packets into the low-priority class. Thus, the average channel coding rate in this case is 0.608 bits/cu, approximately equal to the channel coding rate of 0.60 bits/cu

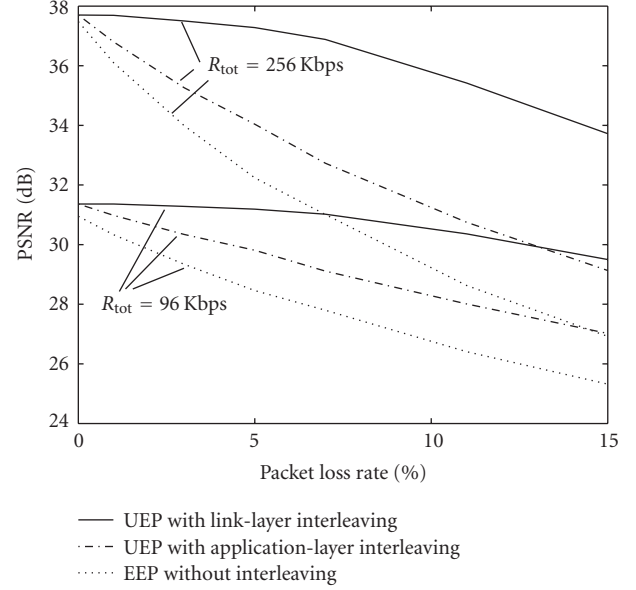


FIGURE 8: Proposed UEP scheme versus EEP scheme; $L_B = 3$; the Foreman sequence.

resulting from using the FEC/EEP scheme employing the fixed RS(5, 3) code.

The results in Figure 8 demonstrate the effectiveness of the proposed approach. By using the FEC/UEP scheme, lower-priority classes are provided with lower-level FEC protection, while higher-priority classes are provided with higher-level FEC protection since the packet losses of the low-priority class will contribute smaller total distortion compared to the packet losses of the high-priority class. Thus, an efficient way of distributing the FEC redundancy over different classes can be achieved. Over a burst-loss channel, the use of our proposed scheme can also randomize the burst errors due to the use of the intraframe interleaving. Specifically, with the use of application-layer interleaving, improved effectiveness of the FEC can be obtained resulting in better reconstructed video quality. Moreover, if link-layer interleaving is used, it substantially improves the performance of FEC coding, resulting in even better reconstructed video quality compared to the application-layer interleaving scheme. This again demonstrates the advantage of using a cross-layer design approach cutting across the application, network, and link layers in order to provide improved quality for video services over bursty packet-loss wireless IP networks.

More specifically, the proposed UEP scheme with link-layer interleaving substantially outperforms both the EEP scheme and the UEP scheme with application-layer interleaving. For example, when the packet loss rate is 15%, for $R_{tot} = 256$ Kbps, the FEC/UEP approach with link-layer interleaving can achieve a 7 dB performance gain compared to the FEC/EEP scheme without interleaving and 5 dB performance gain compared to the FEC/UEP scheme with application-layer interleaving.

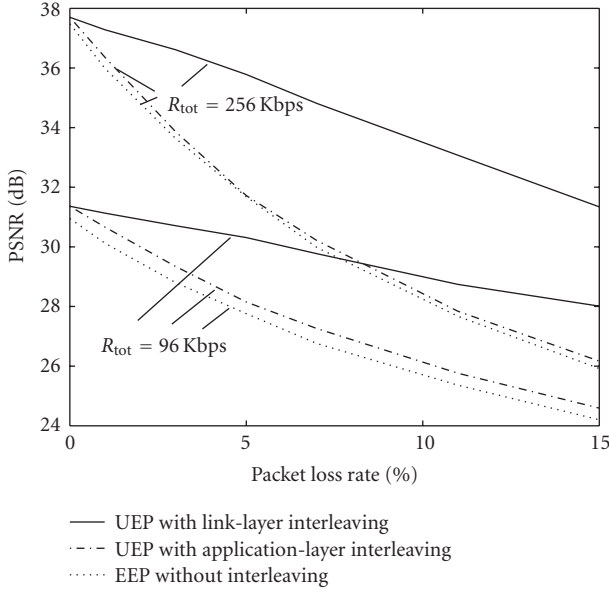


FIGURE 9: Proposed UEP scheme versus EEP scheme; $L_B = 9$; the Foreman sequence.

As demonstrated in [25], the typical burst length in representative UMTS channels is likely to be in the range of 1–3 for application packets with high probability. So, the average burst length at the link layer should be 3–9 for the packetization scheme used in this paper. In order to have a more realistic comparison, we also show the results for the Foreman sequence for link-layer burst length $L_B = 9$ in Figure 9.

In Figure 9, we can see that in this case, the FEC/UEP approach with link-layer intraframe interleaving is still very effective and can achieve, for example, a 5 dB performance gain compared to the other two schemes when the packet loss rate is 15%, for $R_{\text{tot}} = 256$ Kbps. The FEC/UEP approach with application-layer intraframe interleaving can only achieve a very small gain compared to the FEC/EEP scheme, and both systems experience substantially degraded video quality. The reason is that, as described previously, the link-layer interleaving can substantially randomize the burst errors occurring on wireless links, and therefore can make the FEC more effective. On the other hand, as the burst length increases, the use of application-layer interleaving becomes ineffective in dealing with the burst losses that can cause more and more slices to get lost in bursts.

In order to further evaluate our proposed scheme, we repeat the simulations for the QCIF Susie sequence that has a much lower overall motion level than the Foreman sequence. The corresponding results are illustrated in Figures 10 and 11, again for $L_B = 3$ and 9, respectively.

For the results illustrated in Figure 10, for the low-motion Susie sequence, we observe that the proposed UEP scheme with link-layer interleaving still achieves a much higher performance than either the FEC/EEP scheme or the UEP scheme with application-layer interleaving. For example, at $R_{\text{tot}} = 256$ Kbps, when the packet loss rate is 15%, the gain is about 4.5 dB compared to the FEC/EEP scheme and 2 dB compared to FEC/UEP with application-layer

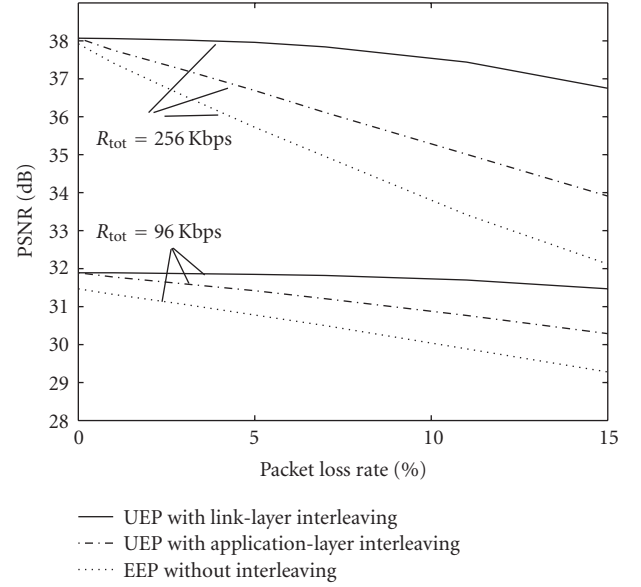


FIGURE 10: Proposed UEP scheme versus EEP scheme; $L_B = 3$; the Susie sequence.

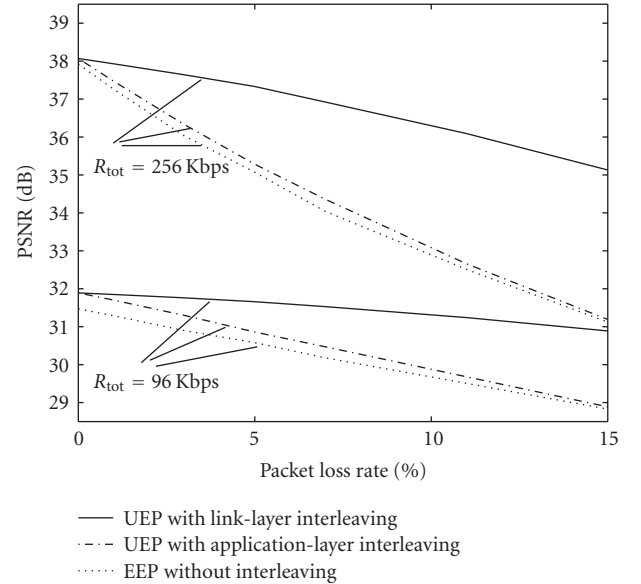


FIGURE 11: Proposed UEP scheme versus EEP scheme; $L_B = 9$; the Susie sequence.

interleaving. Again, the results demonstrate the effectiveness of our proposed approach of employing FEC/UEP together with link-layer interleaving compared to either the FEC/EEP scheme or the FEC/UEP scheme with application-layer interleaving. We should also note that for the low-motion Susie sequence, the gain achieved by the proposed approach is less than that for the high-motion Foreman sequence. The reason is that, as indicated in Figure 3, since the overall motion level of Susie is much lower than that of Foreman, the built-in FEC by itself is very effective in dealing with the packet errors in

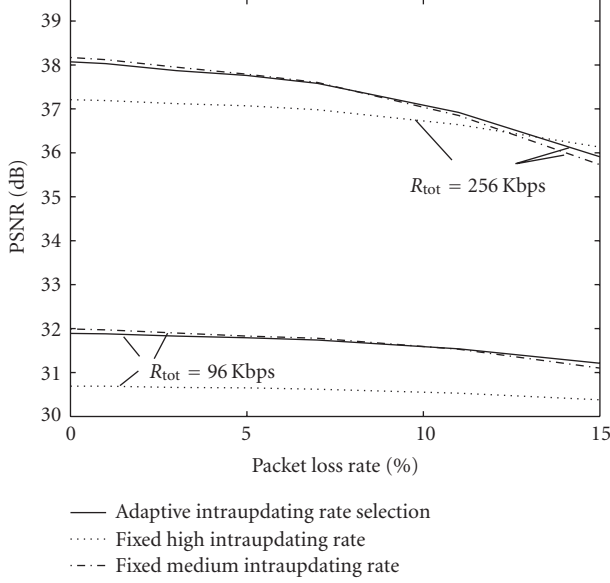


FIGURE 12: Adaptive intraupdating rate selection versus fixed rates; using FEC/UEP with link-layer intraframe interleaving; $L_B = 6$; the Susie sequence.

the Susie sequence. As a result, the FEC coding gain in the case of low-motion sequences is not as much a factor on the reconstructed video quality as in the case of higher-motion sequences.

In Figure 11 for $L_B = 9$, we can see similar performance as shown in Figure 10. Again, we observe that the proposed UEP scheme with link-layer interleaving still achieves a much higher performance than either the FEC/EEP scheme or the FEC/UEP scheme with application-layer interleaving.

Up to this point all results have been for the case of adaptive intraupdating rate selection used in combination with FEC. We now show the relative performance achieved by the adaptive scheme compared to the use of fixed intraupdating rates. In Figures 12 and 13, we illustrate a comparison between using adaptive intraupdating rate selection and using fixed intraupdating rates for the QCIF Susie sequence and the QCIF Foreman sequence, respectively. Here we make use of the proposed FEC/UEP approach with link-layer interleaving and assume a Gilbert channel model with $L_B = 6$. The adaptive intraupdating rate selection is based on EWMA estimation as described previously.

From Figure 12, we see that using arbitrarily fixed intraupdating rates would cause some problems. For example, using a fixed high intraupdating rate can improve the performance for the case of high packet loss rates where P_L is greater than say 13% for both $R_{\text{tot}} = 96$ Kbps and $R_{\text{tot}} = 256$ Kbps. However, this is achieved at the cost of a substantial performance degradation for low packet loss rates. On the other hand, using a fixed medium intraupdating rate can improve the performance for the case of low packet loss rates, but will cause a slight performance degradation when the packet loss rate is high, for example, where P_L is greater than 12%. As illustrated in Figure 12, the adaptive

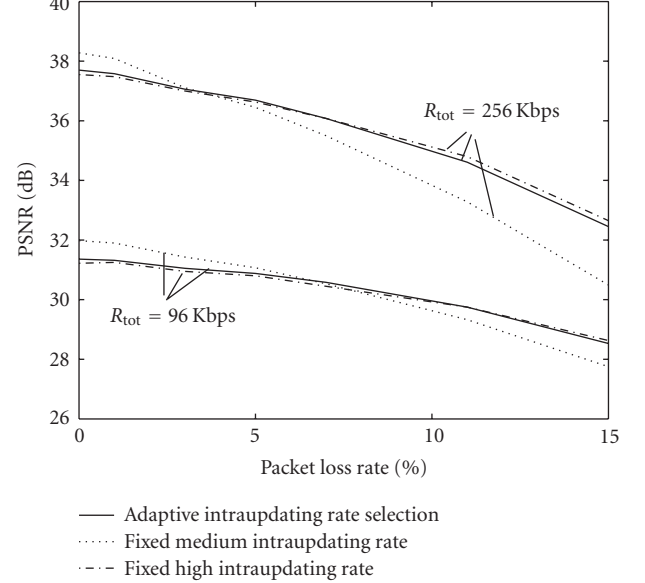


FIGURE 13: Adaptive intraupdating rate selection versus fixed rates; using FEC/UEP with link-layer intraframe interleaving; $L_B = 6$; the Foreman sequence.

scheme can achieve a robust compromise between a fixed high intraupdating rate and a fixed medium intraupdating rate based on the analysis of the motion level of the video sequence. The corresponding performance achieved by the proposed source-adaptive scheme is universally near optimum and makes use only of easily obtained motion information. In Figure 13, we can observe similar performance for the case of the Foreman sequence.

It should be emphasized at this point that the proposed approach is a pure source-adaptive approach that exploits only source information. As a result, as evidenced in Figures 12 and 13, the adaptive approach can be slightly worse than a specified fixed intraupdating rate. But generally, the adaptive approach can achieve a robust compromise between different fixed intraupdating rates based on the motion information of each N contiguous video frames and can achieve near-optimum performance universally for video sequences that are composed of video clips with different motion levels. However, potential performance improvement can be expected if channel feedback adaptation and source adaptation are integrated together. Despite this expectation, it should be noted that in [1] we demonstrated that imperfect channel feedback could substantially degrade the system performance, and in the face of imperfect channel feedback, a pure source-adaptive system may be more appropriate.

The above objective results are based on a quantitative assessment of reconstructed PSNR values. We also show some subjective results based on the reconstructed frames taken from the decoded test sequences. In Figure 14, on the top we illustrate three frames taken from the reconstructed Foreman sequence, while at the bottom the three frames are from the reconstructed Susie sequence, both at $P_L = 7\%$ and $L_B = 9$ with bit rate equal to 256 Kbps. The figures support



(a) PSNR = 30.43 dB.



(b) PSNR = 32.06 dB.



(c) PSNR = 37.22 dB.



(d) PSNR = 34.11 dB.



(e) PSNR = 34.41 dB.



(f) PSNR = 37.11 dB.

FIGURE 14: Comparison of decoded frames for the Foreman and Susie sequences, at the 43rd frame and the 61st frame, respectively: $R_{\text{tot}} = 256$ Kbps, with channel conditions of $P_L = 7\%$, $L_B = 9$; adaptive intraupdating rate selection is used for all results; (a), (d) FEC/EEP scheme; (b), (e) FEC/UEP scheme with application layer interleaving; (c), (f) FEC/UEP scheme with link-layer interleaving.

the preceding objective assessments. Specifically, for both the QCIF Foreman and QCIF Susie sequences, the source-adaptive FEC/UEP approach with link-layer interleaving can achieve the best reconstructed video quality compared to that achieved by either the FEC/EEP scheme without interleaving or the FEC/UEP approach with application-layer interleaving.

4. CONCLUSIONS

In this paper, we have proposed a novel error-resilience framework for video transmission over wireless networks where the channel state information is not available or cannot be obtained easily or accurately. The proposed system is able to adapt to the source motion information inherent in a transmitted video sequence and based on this source-adaptation, a cross-layer FEC-based UEP scheme and an intrarate selection scheme are investigated. Furthermore, within this framework a novel packetization scheme and a novel intraframe interleaving scheme are also investigated. The simulation results demonstrate that this proposed system framework is very effective in dealing with the bursty packet losses occurring on wireless links.

Again, we should point out that this cross-layer approach is applicable on a link-by-link basis that requires the communication parties to be directly connected by a virtual link. If intermediate nodes exist between the sender and the

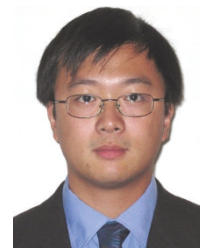
destination, such as in the scenario of the Internet, in order to implement this approach, some special support needs to be provisioned, such as “IP tunneling.”

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